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From the bit to multi-carrier modulation - basics of digital modulation techniques

From a lecture at the VHF conference in Weinheim/Bensheim 2006

Digital modulation techniques have been used in radio engineering for a long time. Together with the use of signal processing it can combine the advantages of digital transmission at low cost. The following overview of digital modulation techniques in the time and frequency domain gives an insight into modern multi-carrier techniques. As an example Digital Radio Mondiale (DRM) is digital short wave broadcasting.

1.

Basic techniques of digital transmission

How does the bit become the RF carrier? The three standard practices of amplitude, frequency and phase modulation can be transferred directly to the digital world. In the simplest form modulation methods that contain only one bit are used, e.g. radio telegraphy using two frequencies for zero and one. If the transmission path is errors free a modulation method can contain more of than two conditions and transfer more than one bit; there are communication techniques with 16 frequencies meaning that the symbol contains 4 bits.

Amplitude Shift Keying (ASK):

The simplest form of amplitude modulation is pulsing the carrier for telegraphy. Since the zero condition cannot be differentiated from an interrupted connection, in practice two amplitude values are used: During transmission of DCF77 for example, 100% and 25% amplitude are used.

Advantages:

- Modulation spectrum is only upward and symmetrical to the carrier frequency
- Modulation can take place somewhere between the oscillator and the antenna
- Amplitude can be directly measured at the receiver

Disadvantages:

- Amplitude errors on the transmission (fluctuations in level, malfunctions) produce errors
- Frequency Shift Keying (FSK): The modulation method uses different frequencies, e.g. RTTY and packet radio use two frequencies for zero and one. It can be interpret either as two phase amplitude modulated channels or as frequency modulation with a non-existent centre frequency with a deviation of ΔF. If the spacing of

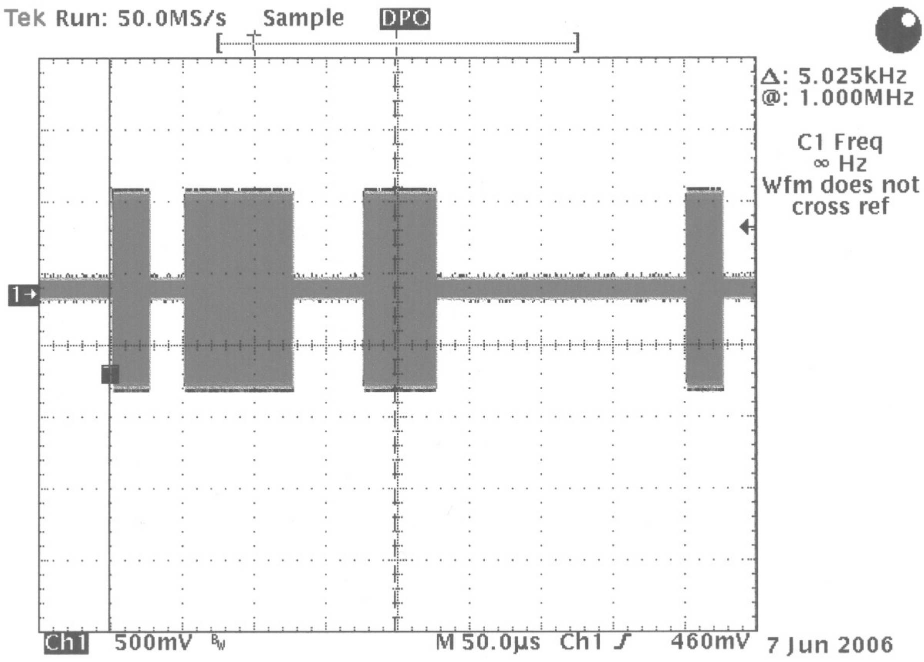

Fig 1: An amplitude modulated 10MHz signal.

the two frequencies is equal to half the data rate, then it is called Minimum Shift Keying (MSK). This is the standard modulation for the GSM mobile phone system.

Advantages:

- Amplitude does not contain any information therefore it is interference proof
- No linear transmitter amplifiers or linear receivers necessary
- Frequency can be directly measured at the receiver

Disadvantages:

- The frequency modulated carrier is theoretically infinitely wide, broader than AM ($B \approx 2\Delta + 2f_{mod}$)
- Modulation requires modulating the oscillator thus worse frequency stability

Phase Shift Keying (PSK):

Using mathematics it can be shown that frequency and phase modulation are very similar and can be converted to one another using suitable filters. The most common method uses two-phase shift keying (0/180°, Binary Phase Shift Keying) and a four-phase shift keying (0/90/180/270°, Quadrature Phase Shift Keying). The two-phase shift keying can be treated as a special case of the amplitude modulation where the carrier frequency is first multiplied by a factor +1 and then by a factor -1. The four-phase shift keying can be considered as two BPSK signals with carriers that are phase shifted by 90°. It can be shown mathematically that these two equivalent carriers do not affect each other and they can be used for BPSK or AM and modulated with different

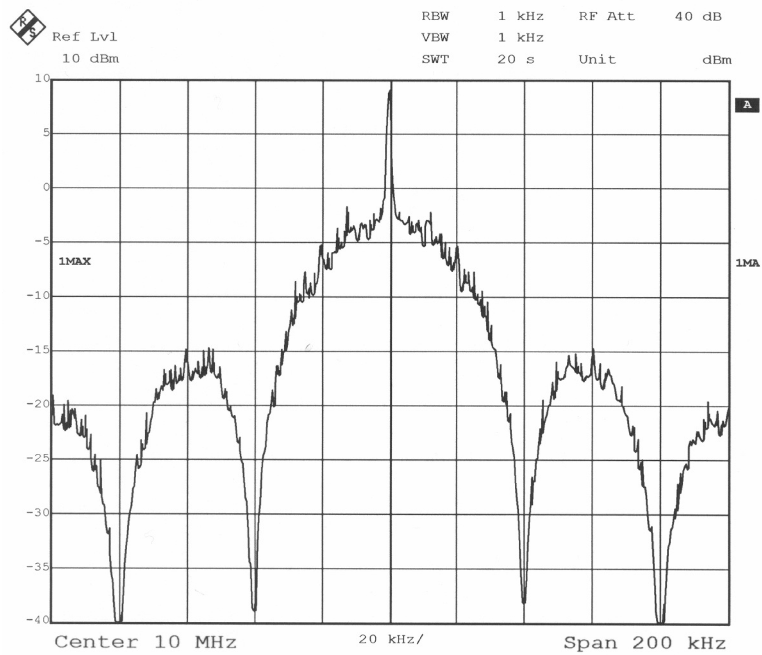

Fig 2: The spectrum of an amplitude modulated signal.

information.

Advantages:

- The phase modulator comes after the oscillator
- Amplitude does not contain any information therefore it is interference proof
- No linear transmitter amplifiers or linear receivers necessary

Disadvantages:

- Phase cannot be directly measured at the receiver, only phase changes
- The frequency spectrum is wider than the comparable AM or FM signals

Combined methods :

Quadrature Amplitude Modulation (QAM): If a transmit path is linear and has a good signal-to-noise ratio, then the data rate can be increased by a combination of amplitude and phase modulation: The amplitude and phase of the carrier is changed at the same time in small steps. Quadrature amplitude modulation accommodates 4 to 15 bits. QAM with 16 states (4 bits) and 64 states (6 bits) is in common use.

The main application for higher QAM methods with 1024 to 32768 states are modems and DSL technologies on fixed line connections.

Advantages:

- Best utilisation of the transmit path in terms of bits/Hz .

Disadvantages:

- Good signal-to-noise ratio necessary
- Linear transmit path from the transmitter to the receiver necessary

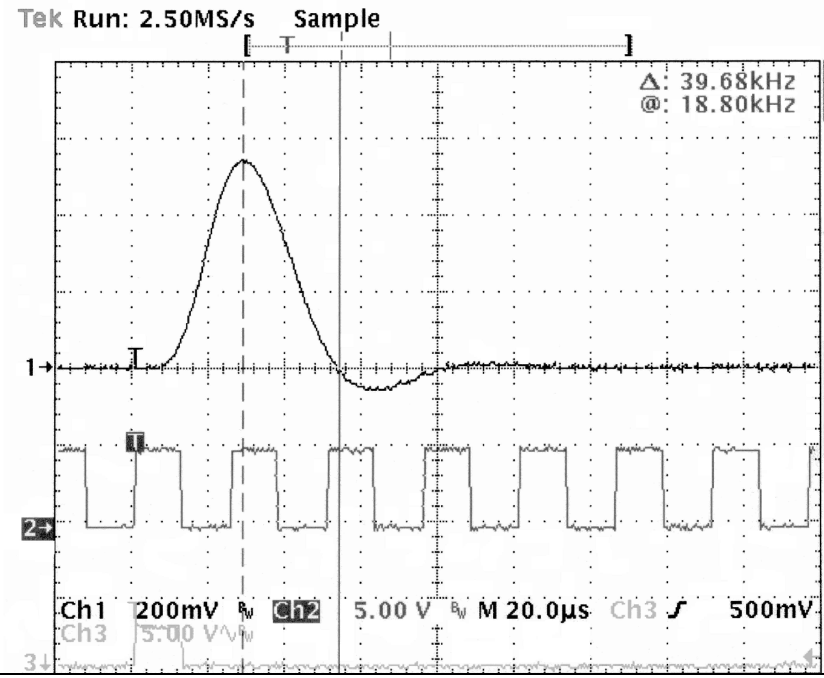

Fig 3: The impulse response of a 19kHz cutoff, 3rd order low-pass Butterworth filter fed with a 40kBit/sec data stream.

2. Modulation in the time domain and frequency domain and filtering of the data signal

It is known that switching the carrier of a telegraph transmitter produces clicking disturbances in neighbouring radio links. A low-pass filter in the transmitter solves the problem if the modulator and the following transmit amplifier are linear. How is the low-pass designed? Harry Nyquist found the answer years ago with the investigation of machine produced telegraph signals:

The most unfavourable case is the bit pattern (01010101….) corresponding to a symmetrical square wave pulse. A data stream of 1kBit/sec represents a square wave pulse with a frequency of 500Hz

If the base frequency of the square wave is fitted to the transmission path the harmonics are filtered out. The data stream then becomes a sine wave and it is difficult to find the centre of the bits, which is a problem for clock regeneration.

Broadband modulation:

A digital data stream has a very fast rise time (74HC logic: Rise time < 10nS), its spectrum extends far beyond data rate. An un-filtered data signal should not be fed to a modulator because it produces broadband modulation that makes neighbouring radio links useless. The same thing happens if an SSB transmitter is over driven producing a square signal feeding the output stage, some contest stations can be heard +/- 300kHz and

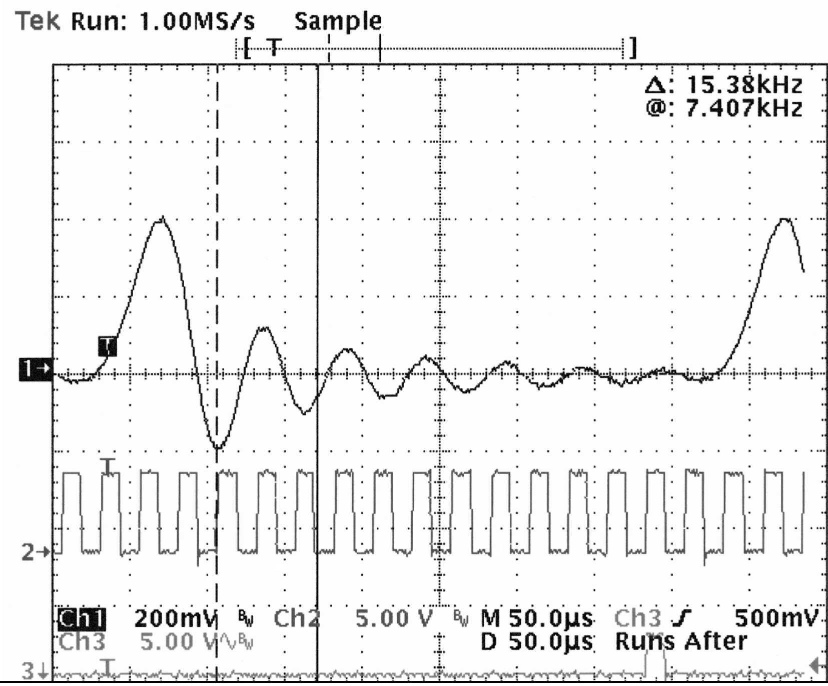

Fig 4: The impulse response of a 19kHz cutoff, 10th order, super-steep, low pass Tschebyscheff filter fed with a 40kBit/sec data stream.

that is not because of a bad receiver!

An amplitude modulator is used for the following examples. A diode ring mixer is used as an AM modulator for a 10MHz carrier. The data stream has a data rate of 40kBit/sec and a 32767 ($2^{15}-1$) bit pseudo random generator is used to represent the data.

The modulated signal is shown in the time domain in Fig 1, the bit length is 25µsec from the 40kBit/sec data rate. The associated spectrum is shown in Fig 2; the carrier frequency is in the centre and the sidebands extend 200kHz to the left and right. For those interested in the mathematics: the amplitude of the sidebands follow (sin x)/x function. With such a signal the neighbouring radio links would be substantial disturbed. The modulation signal must be filtered in a suitable way! Subjectively it is assumed that with reduced rise time of the data signal the range occupied becomes smaller. At first sight a sharp low-pass filter is the solution, but this is discussed further.

Correct filtering of the modulation signal: Most IF bandpass filters in amateur radio equipment are optimised for a rectangular transmission curve, because the Shape Factor is an important advertising feature but most special CW filters under 250Hz ring and are hardly useful. A CW signal only occupies a few Hz so why doesn't it fit in a 100Hz filter? Then there are the strange transmission curves of the IF bandpass filters in a spectrum analyser.

This is where filter theory helps:

If a low-pass filter is fed with a short rectangular voltage pulse or a short RF

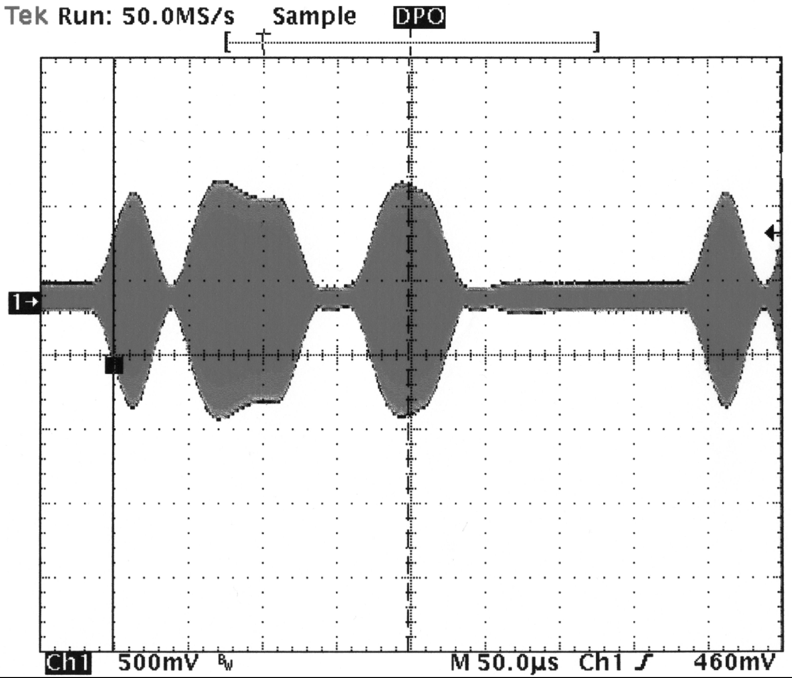

Fig 5: An amplitude modulated signal filtered with a Butterworth low-pass filter.

pulse at the centre frequency of a bandpass filter, then the output is not only a reduced version of the original input but also echoes that are the ringing of the CW filter. The "better" the filter the more echoes can be seen. The impulse response of a 19kHz cutoff, 3rd order low-pass Butterworth filter fed with a 40kBit/sec data stream is shown in Fig 3. The second trace is the clock. The pulse has been beautifully "rounded":

One bit time later the pulse response crosses zero, which means that neighbouring bits in the bit stream are not affected

A negative echo of the origin pulse can be seen, its amplitude is approximately 10 times smaller than the first pulse.

Fig 4 shows a 10th order, super-steep, low pass Tschebyscheff filter with an almost rectangular transmission curve. The data rate is again 40kBit/sec and the cutoff frequency 19kHz. 12 echoes of the origin pulse can be seen fading away sinusoidally so this filter is not useful. The following requirements result for filtering digital signals:

- A low pass filter with a cutoff frequency of half the data rate can be used to filter digital signals and leave the bits recognisable.
- Low-pass filters must have a gradual transition into the cutoff area so that the filter does not ring. Theoretically the optimal solutions are Gaussian filters (as used for IF bandpass filters in spectrum analysers) and Cosine roll off filters with a cosine shaped filter curve.
- The 3rd order Butterworth low pass filter represents a good approximation.
- By small variations to the filter

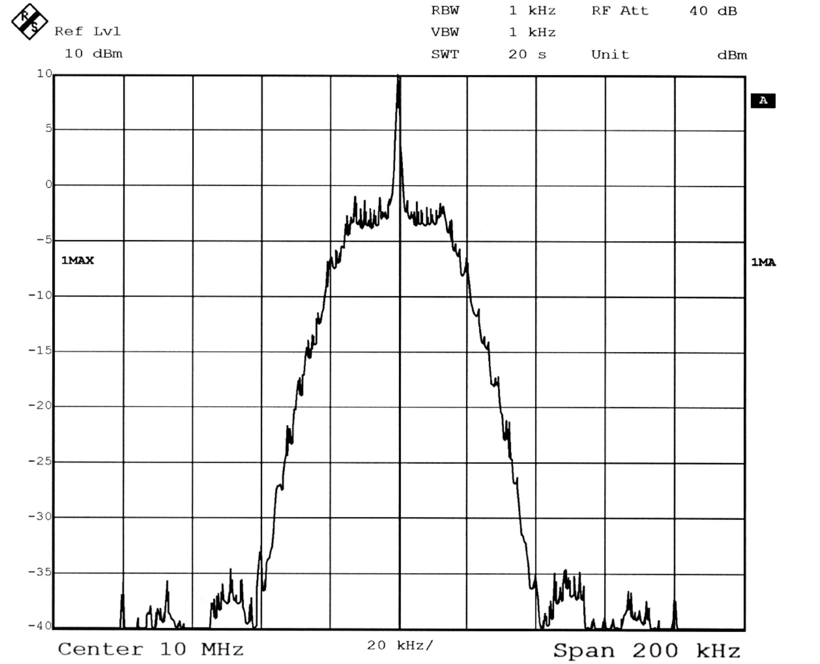

Fig 6: The spectrum of an amplitude modulates signal filtered with a Butterworth low-pass filter.

cutoff frequency the pulse response of the filter can be designed to make the zero cross over in the centre of the following bit so that neighbouring bits affect each other as little as possible.

The advantage of the last point can only be achieved if the bit timing is recovered in the correct phase at the receiver and the bit stream is scanned in the centre of the bit timing. This highlights two additional tasks: The transmitter must send sufficient bit changes independently of the bit stream to ensure that the bit timing can be recovered. The receiver must be able to recover the bit timing, usually using a PLL circuit.

The success of filtering with a Butterworth filter is show in Fig 5 in the time domain and Fig 6 in the frequency domain. The formerly rectangular RF envelope is rounded and the power density at more than 40kHz from the carrier is practically zero.

Disturbed transmission paths:

Digital transmission techniques were introduced with the goals of improving quality and of reducing the equipment cost. To start with digital transmission techniques were concerned with the increase of the bit rates into the Gigabit region so that the gigantic capacity of fibre optic cable could be used. That can be achieved with carefully planned transmission paths that have a good, well-behaved and smooth frequency response and low noise. These conditions are fulfilled by cable but not by radio paths.

- On long cable runs the characteristic impedance jumps if the connections are not implemented correctly causing humps in the frequency response.
- Multi-path propagation in a hop

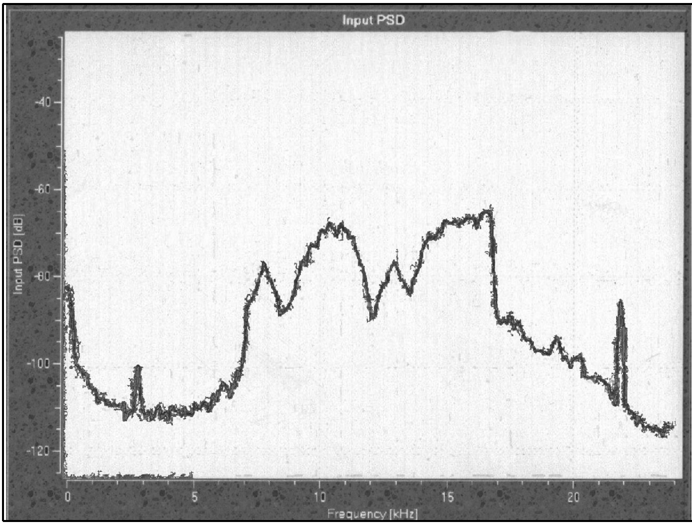

Fig 7: Frequency responses of a DRM transmission on short wave

produces sharp "holes" in the frequency response caused by interference

- Foreign transmissions disturb both cable runs (cross talk from other cable cores or unsatisfactory shielding) and radio links.
- The frequency response of DRM reception taken with strongly disturbed frequency is shown in Fig 7 and Fig 8.

The following is applicable:

- A frequency response equaliser gives good results so that the frequency response is not too uneven.
- Some radio systems use automatic channel selection and look a good radio link
- Using frequency diversity technology the information is transmitted on two frequencies in

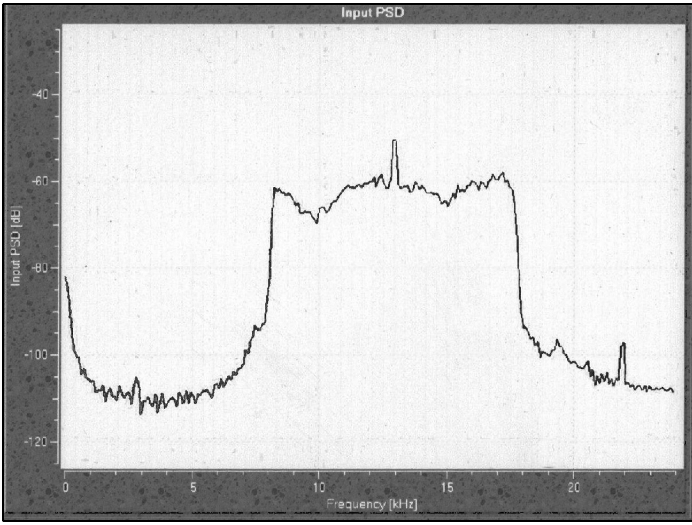

Fig 8: Frequency responses of a DRM transmission on short wave

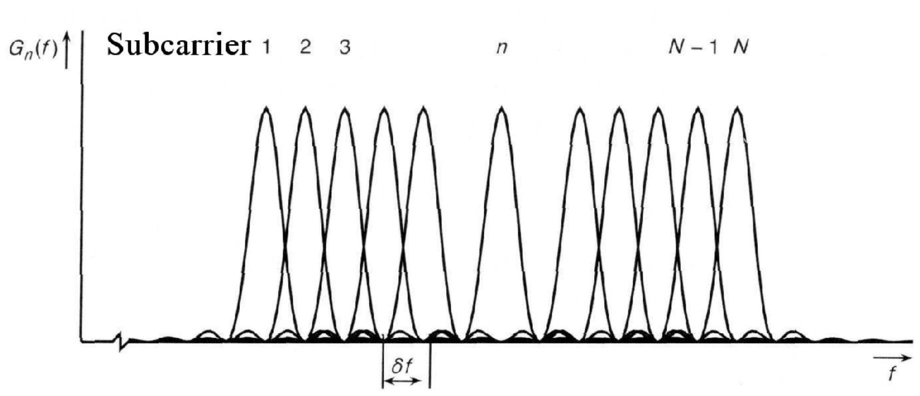

Fig 9: Carriers for an orthogonal frequency multiplexed system (OFDM).

the hope that one is good.

Area diversity uses a second receiving antenna shifted by some wavelengths that often supplies unimpaired reception during multi-path propagation.

A difficult radio link in California has used double frequency diversity and quadruple area diversity for some years.

Advantages of multi-carrier modulation:

The frequency diversity method can be used with more consistency. Before the digital age telephone calls were converted using frequency-multiplexing equipment into single sideband modulation; the method meant that the calls could be transmitted using both cable and radio (radio relay link or satellite communication). If a sub channel was disrupted it was not used. With a disrupted channel unused only the transmission rate was affected and the connection was not broken, the same applies to a digital transmission.

This advantage has been rediscovered with the availability of special signal processors thus overcoming the arguments against frequency division multiplexing. The modulators and filters for each sub channel can be realised digitally using cheap signal processors. There are many cable and radio systems using digital frequency multiplexing equipment:

- ADSL technology (Asymmetric Digital Subscriber Line) uses 4kHz wide sub channels with 4.3125 kHz spacing
- DAB (Digital Audio Broadcasting) uses 1536 sub channels in a 1.5MHz spread spectrum
- DVB-T (Digital Video Broadcasting - Terestrial) uses 2048 or 8192 sub channels in one 7MHz wide Television channel
- DRM (Digital Radio Mondiale) uses the 9kHz wide AM radio channels with 40 to 110Hz(!) spread sub channels.

Even the portable radio planners think after the failure of UMTS and spreading spectrum modulation multi-carrier modulation (3G) that this is the 4th generation portable radio system.

It must not be forgotten that a multi-carrier modulation system has a considerable administration overhead. With a two-way system users can still operate with the loss of a single channel of the distributed channels. Broadcast systems send enough redundant

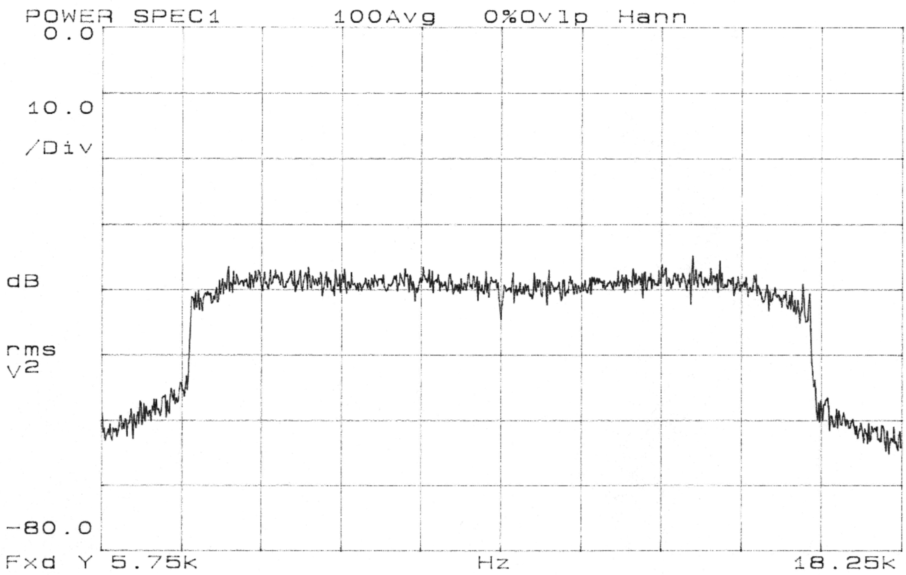

Fig 10: Spectrum of a DRM signal as seen on a spectrum analyser.

information so that it does not present a problem if a certain portions of the channels are not usable.

A recognised disadvantage of multi-carrier systems is the long synchronisation time: a DSL modem does not need many seconds to synchronise, depending upon line quality, a transmitter search can be used for digital television broadcasts. The developments to a DRM similar amateur radio procedure run however!

Summary of the pros and cons of multi-carrier systems:

Advantages:

- Well suited to bad transmission paths with poor frequency response or sharp breaks
- The frequency response only needs to be good in the sub channel
- Data with a low rate is transferred in the sub channel. Thus the transmission becomes insensitive to multi-path propagation. The requirement is that the transit time difference between the multi path waves at the receiver must be smaller than the symbol length
- Individual sub channels can fail completely during transmission

Disadvantages:

- The modulation is generated with considerable amounts of digital signal processors
- Pilot channels must be inserted for recovery
- The synchronisation procedure is long therefore it is suitable for permanent connections

Practical realization: DMT and OFDM:

The term Digital Multi-carrier Modulation (DMT) is self-descriptive.

The term Orthogonal Frequency Division Multiplex (OFDM) is also used in connection with multi-carrier modulation. Orthogonal means right-angled, but why is this used for a transmission technique? Sine and cosine waves are orthogonal for the same frequency, if they are multiplied together then the dc component is zero. That has been used for many years in quadrature amplitude modulation for PAL colour television transmissions to transfer two channels on a carrier.

The frequency spacing of the sub carriers for a multi-carrier modulation can be chosen freely as long as the spectra do not overlap themselves. There are carrier spacings where the mutual disturbance becomes minimal thus saving on filtering.

Looking again at Fig 2: Although the data signal is not filtered, the spectrum falls to zero symmetrically around the carrier at a spacing of the bit rate. Therefore it is best to select the frequency spacing of the sub carriers at integer multiples of the symbol rate. The first case (standard OFDM) does not give the closest packing of the sub carriers but theoretically the sub channels do interfere. This is because the information is carried in the range of the carrier frequency plus/minus half the symbol rate as shown in Fig 9 from [1]. This cannot be seen on a spectrum analyser; only the rectangular form of the whole spectrum can be seen as shown in Fig 10.

3. Digitally Radio Mondiale - current state

In the last few years some digital broadcast and television systems have been introduced, whose use for the ordinary consumer is either doubtful or who does not fulfil expectations created by advertising:

- Digital Satellite Radio (DSR) failed after 5 years due of lack of participants!
- Digital audio Broadcasting (DAB) has been operating for years in the general market and should replace VHF FM broadcasts. The quality loss compared to VHF FM is not great but it is doubtful if the population will accept 100 million worthless radios in Germany alone even after a long period of use and low priced receivers. It is not clear what the thoughts of DAB are in other countries. (In the UK it is even more troublesome because the BBC standard for DAB is not the same as the European standard and is due to be harmonised with even more useless receivers – Andy)
- Digital Video Broadcasting (DVB) was publicised to have better image quality and in addition still more programs. These are (usually) of the lowest technical quality (data rate under 2MBit/sec) and squeezed into the transmission paths. With modern large screen televisions one the bad image quality can be seen showing “dancing pixels”.
- DVB-T has the advertising slogan of television for all, many programs using a Yagi antenns..
- DVB-H (television on the mobile phone), it is doubtful if this will become a market success.
- Now DRM (Digital Radio Mondiale) is about to digitise the medium and short wave broadcast bands. Is a market success to be expected here? No mass market, but an interesting addition: a clear jump in audio quality (almost VHF FM quality) compared to AM broadcasts and at the same time worldwide reception with only telescopic antenna an no additional infrastructure.

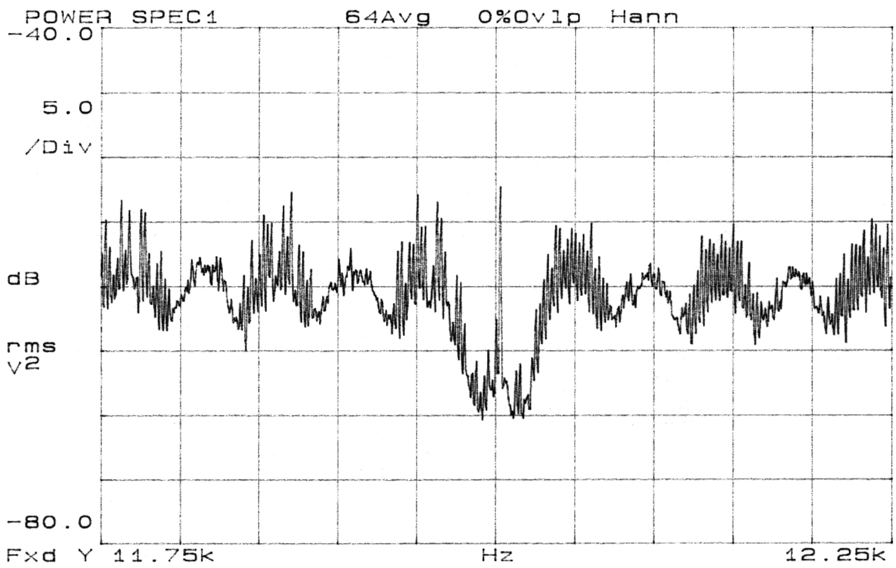

Fig 11: Spectrum of a DRM signal around the carrier centre (high resolution).

3.1. DRM has the following characteristics

- The 9 or 10kHz channels used on the long, medium, and shortwave bands are sufficient
- Further use of existing, modern transmitting plants
- Good reception with low signal/signal-to-noise ratio
- Stability against multi-path propagation in the ionosphere
- Convenient operation as with modern VHF FM receivers with Radio Data System stem (RDS)

In addition the following benefits are:

- A new, efficient audio data compression technique called Adaptive Audio Coding with High Quality Spectral Band Replication (AAC with HQ SBR) supplies VHF FM quality with a data rate around 25kBit/sec.
- The transmission uses digital frequency multiplexing (Orthogonal Frequency Division Multiplex) and splits up the 9kHz channel into 40 to 110Hz wide sub channels that are modulated with a low data rate. Depending upon the system the sub channels are modulated with four-phase quadrature amplitude modulation (2 bits/symbol) up to 64-phase (6 bits/symbol). The long symbol duration makes the system insensitively to multi-path propagation; the multi-carrier procedure has sufficient redundancy that the loss of a carrier by Fading or disturbers can be corrected.
- Interleaving of the data over several 100 milliseconds improves the error correction in the case of bundle errors.

The disadvantage of the system is that despite special synchronisation channels, synchronisation takes a very long time.

Table 1: DRM Modes of error protection.

Mode	Duration Tu spacing 1/Tu Tg	Carrier guard interval Ts=Tu+Tg	Duration of symbol frame Ns	Duration of symbols per	Tg/Tu	Number
A	29mS	41 2/3Hz	2.66mS	26.66mS	1/9	15
B	21.33mS	46 7/8Hz	5.33mS	26.66mS	1/4	15
C	14.66mS	68 2/11	5.33mS	20mS	4/11	20
D	9.3mS	107 1/7Hz	7.33mS	16.66mS	11/14	24

The following points affect this:

- Search of the pilot channels, carrier synchronisation and determination of the mode
- Waiting for the Interleave period
- Error correction and data expansion

The entire procedure takes some seconds, so that it is not possible to carry out a transmitter search in the usual sense. Because modern world receivers have a stored frequency lists and transmission timetables stored in EEPROMs, an automatic search of the data channels in the DRM system could be the solution.

Digital Radio Mondiale cannot replace the normal AM broadcasts. In addition the existence of a billion AM receivers is too large. It will be an interesting alternative for holidaymakers to hear their home stations on a telescopic antenna without systems such as the Internet or a satellite antenna. Only current saving DSP demodulators and AAC audio decoder are missing. So far there are unfortunately only expensive receivers. Phillips have announced a DRM demodulator IC for car radios!

3.2. The DRM modulation format

Channel width:

Standard 9kHz (LW, MW) and 1 kHz (SW), in addition there are half channels with 4.5 or 5kHz spacing and double channels with 18 or 20kHz spacing.

Stages of the error protection (Robustness):

There are 4 stages A to D (Table 1); A is only for channels with Gaussian distributed noise to D for channels with time-variable multi-path propagation with high transit time differences and Doppler shift.

Logical channels:

The Fast Access Channel (FAC) is the synchronisation channel; the Service Description Channel (SDC) is for decoding the multiplexer sub data streams, the Main Service Channel (MSC) contains the data.

Symbol duration:

16 2/3mS, 20mS, 26 2/3mS

Transmission capacity for data:

Depending upon channel width and error protection, 6 to 55kBit/sec (theoretical range 4.8 to 72kBit/sec), 15 to 25kBit/sec is usually used.

Pilot channels:

750Hz, 2250Hz and 3000Hz above the channel cutoff frequency

Signal-to-noise ratio required:

15 to 23dB depending upon channel model (only noise up to multi-path propagation with 6mS transit time difference and 4Hz of Doppler shift), under favourable conditions 10 to 11dB

Data compression audio signal

AAC (Adaptive Audio Coding), optional SBR (Spectral Band Replication) production “synthetic level”, alternatively mono or Stereo

4. Literature:

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[2] DVB digital TV, U Freyer, Verlag Technik 1997

[3] The basics of the television engineering, G. Meals , Springer Verlag 2005

[4] ETSI Technical Specification TS 101 980. ver. 1.1.1 (September 2001)

[5] BBC R&D White paper WHP 064, Digitally Radio Mondiale - revitalising the bands below 30 MHz, July 2003

The UK Six Metre Group

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With over 700 members world-wide, the **UK Six Metre Group** is the world's largest organisation devoted to 50MHz. The ambition of the group, through the medium of its 56-page quarterly newsletter **'Six News'** and through its web site **www.uksmg.com**, is to provide the best information available on all aspects of the band: including DX news and reports, beacon news, propagation & technical articles, six-metre equipment reviews, DXpedition news and technical articles.

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Michael Gabis, Ralf Rudersdorfer

Current digital radio standards similar to FM voice transmission, Part 2

Continued from 2/2008

In the first part of this article the general conditions of digital radio standards were described, in a similar way this article describes the DECT standard. A comparison is made to the similar FM voice transmission standards. The achievable range as well the ability for trouble free reception and reception of data signals at low signal level is considered under comparable conditions.

2.7. DECT

DECT (Digital Enhanced Cordless Telecommunications) [19], [24], [4] was specified as an ETSI standard in 1992. It is a micro-cellular digital portable radio network for high user densities, to be used predominant in company buildings for data and language transfer. A combination of TDMA and FDMA are used for the DECT standard. The available frequency spectrum from 1880MHz to 1900MHz is divided into 10 channels with 1728kHz spacing. The centre frequency f_c is calculated as follows:

$f_c = f_0 - c \bullet 1728\text{kHz}$ with $C = 0, 1,\ldots, 9$

and $f_0 = 1897.344\text{MHz}$ {1}

The centre frequency for an active state can vary by a maximum of ±50kHz. The modulation is Gaussian minimum Shift Keying (GMSK) with a range time product, $B \cdot T = 0.5$. GMSK is a special type of the MSK (Minimum Shift Keying). Instead of switching violently between the two frequencies, the switching edges are flattened using a Gauss filter leading to the fact that the signal range is reduced.

The transmission capacity of each channel is divided into 10ms long frames containing 11520 bits. This framework gives a data transmission rate of 1152kbit/s. Each 10ms long frame is divided into 24 time slots. The first 12 time slots are intended for downlink, the next 12 for the uplink (Fig 6). For a duplex connection pairs of time slots are formed following each other within 5ms. Each of the 24 time slots has a length of 480 bits. The first 32 bits are reserved for the preamble and synchronisation. For a connection 64 bits of control information follow:

C (signalling for higher layers)

P (Paging information)

Q (system information, downlink only)

M (MAC Layer information)

N (handshake, identity information)

as well as 320 bits of useful information and a 4 bit parity check. A protection gap of 60 bits forms the remainder.

DECT uses a building block principle, the system component parts are: the air interface (Physical Layer), the access layer (Medium Access Control layer,

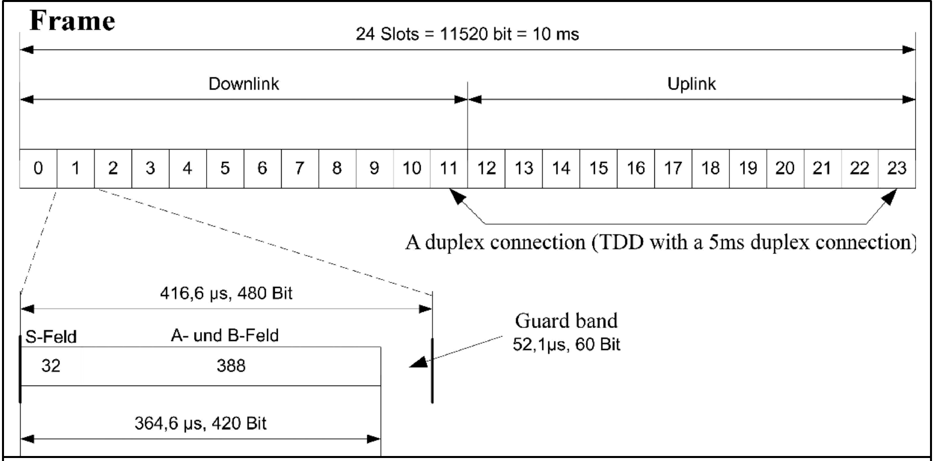

Fig 6: Frame structure for a DECT system.

MAC Layer), the link layer (Data Link Control Layer), the network or switching layer (Network Layer) and the administration of the lower layers (Lower Layer Management Entity).

Changing the radio cell without any interruption passes on a conversation for a moving mobile user. The maximum distance between the base station and mobile equipment in a free range is 300m and in buildings is up to 50m. This range can be increased using relay stations or mobile relay stations. For cost reasons DECT was developed for user speeds up to 20km/h.

The bit duration is 0.868µs. Without error correction time differences of under 10% of the bit duration are permissible, DECT can work with a delay spread up to 200 – 300nS, which corresponds to a transmission distance difference of approximately 100m being suitable for difficult environments Note: Stations with big reflections or problems with reflecting metal walls suitable cell planning and sector antennas can be used if necessary.

DECT offers the following service possibilities:

- Quality speech transmission in fixed network
- Duplex data communication rates with a total bit rate of 38.8kBit/s. In the half duplex mode the total bit rate of up to 931.2kBit/s is possible if all 24 time slots used. Still higher data rates are possible by multiplexing several carriers. Thus many cordless data applications are possible such as Internet, ISDN, Fax or TV.
- Applications for DECT covers PBX (Private Branch Exchange, private PABX), Telepoint and wireless local networks as well as speech telephones.

The DECT system can handle up to 1000 users in a Local Area (LA) [7]. For larger numbers of users, different cell areas can be planned that can be administered internally by DECT. A Dynamic Channel Selection procedure (DCS) is used to manage the high speech and data traffic loads and their uneven load on a cell with temporary and very variable load peaks. Thus in principle the entire frequency spectrum with all 120 channels are available to the mobile station and a suitable channel is selected in each cell. The 120 channels result from the 10 carrier frequencies and the 12 time slots for the downlink or uplink.

In cellular portable radio systems (GSM,

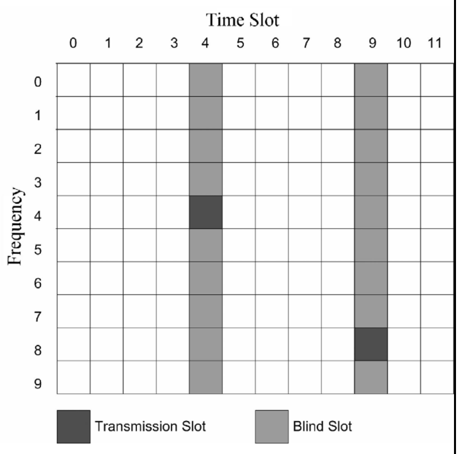

Fig 7: Blind slots for a transceiver with two connections and a permanently installed station.

UMTS,…) channel assignment takes place via a Fixed Channel Allocation (FCA). This is a very precise cell planning system but short load changes on an individual cell are not dealt with very well. With DECT there is no frequency planning, just one planning for the locations is necessary called DCS. The system can lower the blocking probability independently of changing loads adjust itself better than FCA networks. The dynamic assignment takes place according to certain rules so that there are no large disturbances. One rule is the interference adaptive channel assignment; dispatching decisions are made based on online measurements of the interference level. There are two types central and decentralised systems, DECT uses decentralised systems. The highest capacity gains can be obtained with centralised systems because all cell values are available to the control system and all dispatching decisions can be optimised based on the disturbances in adjoining cells. However the signalling expenditure between cells and control unit is higher. With decentralised systems, like with DECT, each permanently cell installed decides independently of other cells, so that the danger exists that channel dispatching will clash with another cell. A high threshold value for the C/I (Carrier to Interference ratio) reduces this danger, however this affects the capacity gains. The advantage is that there is no frequency planning and the system can cope with changes in network loads. Using interference adaptive channel assignment procedures it is possible to extend a DECT system by simply adding further permanently installed stations and increase the capacity.

The actual maximum number of channels of a DECT cell depends on the existing number of transceivers. If it has only one transceiver, then only one mobile station can be served per time slot, since a transceiver cannot work with two different frequencies at the same time. Thus only 12 channels can be used. If a channel on a terminal is occupied by a mobile, then the remaining channels in the same time slot are marked as blind slots (Fig 7). The mobile station looks for the possible channels within each framework in order to attain information about their quality. If the terminal occupies a channel for transmission then it cannot supervise the channels of the other frequencies in this time slot. Because of economies in the design it is not usual to be able to examine the time slots for quality before and after use. The blocked channels are no longer available (Fig 8).

To make the data transmission rate of individual connections variable several time slots can be assigned. This dispatching can take place for the uplink and downlink asymmetrically.

The digital coding used for DECT system is Adaptive Differential Pulse Code Modulation (ADPCM) using 32kBit/s.

In micro-cellular systems with small cell sizes and moving mobile stations the number of cell changes are counted. Because of the substantial signalling overhead it is best to keep the number of handovers as low as possible. The control system uses a decentralised handover algorithm steered by the mobile station

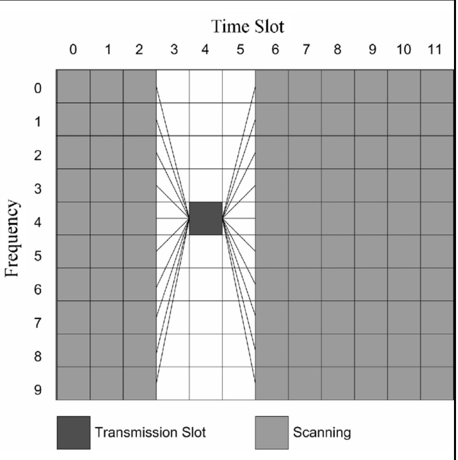

Fig 8: Un-usable channels because of the switching duration of the mobile station channel measurement.

that decide whether and when a handover is necessary. A seamless handover is possible because the old channel is only left if the new channel is already available. The user does not notice anything unlike the non-seamless handover schemes.

In order to protect DECT against unauthorised interference and abuse rules were formulated that ensure a high level of system security:

- Identification of a participant
- Avoidance of unauthorised use of the mobile station
- Identification of a permanently installed station
- Avoidance of unauthorised use of permanently installed stations
- Illegal listening to user or signalling data

The following safety precautions, similar to those used in public portable radio systems, are part of the standard system: A pin enquiry can be implemented when the mobile station is switched on. A mobile station has an international subscriber identification that is clearly defined in a certified range. Each mobile station is examined before the connection is establishment in order to prevent unauthorised network access. In addition an authorising key is stored in the mobile station and in the base station that is compared by an identification procedure. Thus a mobile station is protected against being manipulated by a permanently installed station to access the network. To protect other users from illegal listening to the user or signalling data the radio interface uses coded transmission [24].

3. Comparison and evaluation

A direct comparison of the wireless transmission systems from chapter 2 in comparison to the similar FM voice transmission can made on the basis of the performance of standard receiver sensitivities. This is a measure of the quality of the demodulated signal from the smallest possible RF signal; it also determines the best transmission range. For speech transmission, based on similar types of modulation, the signal quality is expressed as signal/signal-to-noise ratio. For digital transmission the Bit Error Rate (BER) is used to determine the quality. The Bit Error Rate is the relationship of the number of bits that were incorrectly demodulated to the total number of bits sent. A bit error rate of 6 * 10^{-6} means that on the average 6 bits can be wrong if 1 million bits were sent.

A detailed search of the receiver sensitivities in the literature showed that often they couldn't be compared because the Bit Error Rates are defined differently. To be able to compare these systems on the basis of receiver sensitivity, the receiver sensitivities were converted for a Bit Error Rate of 10^{-3}. For DECT no conversions were necessary since the standard Bit Error Rate is referred to 10^{-3}.

3.1. Receiver sensitivities

The BER for an AWGN channel has the

form $$P_b = a \cdot Q\left(b \cdot \sqrt{c \cdot \gamma_b}\right) \quad \{2\}$$

Where constants a, b and c constants depend on the modulation system and the signal to noise ratio

$$\gamma_b = \frac{E_b}{N_0} \quad \{3\}$$

with the signal energy of a bit Eb and the noise performance density N_0. Q () represents the Q-function.

3.1.1. Q-function

The Q-function {4} [15] is often used in communication systems instead of the complementary error function. The probability for an incorrect (BER) is defined with the help of the Q-function.

$$Q(x) = \frac{1}{2} erfc\left(\frac{x}{\sqrt{2}}\right) \quad \{4\}$$

The following are calculated values of the Q-function:

Q(x) = BER	x
10^{-5}	4.26489
10^{-3}	3.09023
2×10^{-3}	2.87816
10^{-2}	2.32635
1.2% = 0.012	2.25713
1.5% = 0.015	2.17009
2.4% = 0.024	1.97737
2 x 0.024	1.66456

3.1.2. Tetrapol

Modulation: GMSK BT = 0.25

Sensitivity is indicated for Tetrapol for a BER of 1.5% [2]. The error probability is calculated according to [16]:

$$P_b = Q\left(\sqrt{\frac{2 \cdot \alpha \cdot E_b}{N_0}}\right) \quad \{5\}$$

a ≈ 0.68 for BT = 0.25

BER = 10^{-3}:

$$3.09.23 = \sqrt{2 \cdot 0.68 \cdot \gamma_{b2}} \Rightarrow \gamma_{b1} = 7.02 \rightarrow 10\log \gamma_{bidB} = 8.46dB$$

BER = 1.5%:

$$2.17009 = \sqrt{2 \cdot 0.68 \cdot \gamma_{b2}} \Rightarrow \gamma_{b2} = 3.46 \rightarrow 10\log \gamma_{b2dB} = 5.39dB$$

Correction factor = $\gamma_{b1dB} - \gamma_{b2dB} = 3.07dB$

3.1.3. GSM/GSM-R/GSM-BOS

Modulation: GMSK BT = 0.3

With GSM sensitivity for a BER of 2.4%, based on a data sheet of the Sony Ericsson K610 mobile telephone. In accordance with [26] the error probability is computed:

$$P_b = 0.5 \cdot Q\left(\sqrt{d^2{}_{\min} \cdot \frac{E_b}{N_0}}\right) \quad \{6\}$$

d^2min = 1.7874 for BT = 0.3

$$P_b = 0.5 \cdot Q\left(\sqrt{d^2{}_{\min} \cdot \frac{E_b}{N_0}}\right) \rightarrow 2 \cdot P_b = Q\left(\sqrt{d^2{}_{\min} \cdot \frac{E_b}{N_0}}\right)$$

BER = 10^{-3}:

$$2.87816 = \sqrt{1.7874 \cdot \gamma_{b1}} \Rightarrow \gamma_{b1} = 4.63 \rightarrow 10\log \gamma_{b1dB} = 6.66dB$$

BER = 2.4%:

$$1.66456 = \sqrt{1.7874 \cdot \lambda_{b2}} \Rightarrow \gamma_{b2} = 1.55 \rightarrow 10\log \gamma_{b2dB} = 1.90dB$$

Correction factor = $\gamma_{b1dB} - \gamma_{b2dB} = 4.75dB$

3.1.4. TETRA

Modulation: p/4-DQPSK (coherent). Sensitivity is defined for TETRA as a BER of 1.2% [1]. The error probability amounts to [16]:

$$P_b = Q\left(\sqrt{2 \cdot \gamma_b}\right) \quad \{7\}$$

BER = 10^{-3}

$$3.09023 = \sqrt{2 \cdot \gamma_{b1}} \Rightarrow \lambda_{b1} = 4.77 \rightarrow 10\log \gamma_{b1dB} = 6.79dB$$

Table 2: System types and receiver sensitivity.

Reference standard	BER= 10^{-3}	20dB	----------------- BER = 10^{-2}	---------------------		
	DECT	FM speech	TETRA	Terapol	DIIS	GSM, GSM-R GSM-BOS
Receiver sensitivity	-94dBm	-115dBm	-104dBm BER= 1.2%	-111dBm BER= 1.5%	-110dBm	-102dBm BER> 2.4%
Modulation	GMSK	FM	p/4 DQPSK	GMSK	CP-4GFSK	GMSK
Correction factor for BER=10^{-3}		+2.7dB		+3.1dB	approx. +2.5dB	+4.8dB
Corrected sensitivity BER=10^{-3}bBm	-94	-115	-101.3	-107.9	-107.5	-97.1

BER = 1.2%:

$$2.25713 = \sqrt{2 \cdot \gamma_{b2}} \Rightarrow \gamma_{b2} \rightarrow 10 \log \gamma_{b2dB} = 4.06dB$$

Correction factor = $\gamma_{b1dB} - \gamma_{b2dB} = 2.73dB$

3.1.5. Interpretation

In Table 2 all systems described in chapter 2 are compared based on receiver sensitivity as described in chapter 3.1. The systems are arranged according to their BER. It can be seen that none of the digital systems approaches the good receiver sensitivity of a similar FM voice transmission. This shows that FM voice transmission has receiver sensitivity better than the state of the art problem free systems. It can be seen that the narrow band systems Tetrapol, TETRA and DIIS exhibit the best receiver sensitivities of the digital systems. This is because the sensitivity of a receiver is directly related to the noise of the received spectrum.

Receiver sensitivity of similar transmission modes is referred to by a certain signal/signal-to-noise ratio of the demodulated signal and often defined as Signal, Noise and Distortion to Noise and Distortion (SINAD). SINAD describes the logarithmic relationship of signal to noise performance in dB, where distortions and all other defects are considered. A further comparison assumes that for a similar voice transmission with a BER of 10^{-3} corresponds to 20dB SINAD.

3.2. Maximum range

The maximum possible attenuation between transmitters and receivers is shown in Table 3. It shows the result of the difference in transmitter power with receiver sensitivity corrected for a BER of 10^{-3}.

$$a_{max} = P_{Smax} - P_{Emin} \quad \{8\}$$

where a_{max} is the maximum attenuation in dB, P_{Smax} is the maximum transmitter power in dBm and P_{Emin} is the receiver sensitivity in dBm. For the case for obstacle-free propagation between the transmitter and receiver

$$a_{max} = a_0 \quad \{9\}$$

with the free apace attenuation a_0dB and thus the maximum range r_{max}

$$r_{max} = \frac{\lambda}{4\pi} \cdot 10^{\frac{a_{max}}{20}} \quad \{10\}$$

in metres when using isotropic antennas.

3.2.1. Interpretation

Table 3 shows what is evident from

Table 3: System comparison on the basis maximum possible range.

	DECT	FM speech or BOS		TETRA	Tetra pol	DIIS	GSM GSM-R GSM-BOS
Corrected sensitivity BER=10^{-3}dBm	-94	-115		-101.3	-107.9	-107.5	-97.2
Permitted transmitter power							
Base station W	0.25	25		25	25	25	25
Base station dBm	24.0	43.98		44	44	44	44
Mobile W	0.25	5		1	1	5	2
Mobile dBm	24	336.99		30	30	37	33
Maximum attenuation between transmitter and receiver							
Base station dB	118	158.98		145.3	151.9	151.5	141.2
Mobile dB	118	151.99		131.3	137.9	144.5	130.3
Maximum range due to free space attenuation							
Used freq. MHz	1890	150	75	390	390	390	900
Wavelength m	0.16	2	4	0.77	0.77	0.77	0.33
Range km	10	6328.6	12657.1	224.1	482.3	1026.4	86.4

Table 2 that none of the digital systems reaches the range of the similar system. The theoretical range for FM voice transmission is high because on the one hand the transmitting power is higher in the comparison to for instance DECT and on the other hand due to the different free space attenuation of the frequency bands used.

For an objective comparison of the systems a more exact analysis is required.

3.3. Comparison for the same frequency and same transmitting power

For this comparison the maximum range was compared using the equation {10} that uses the free space attenuation under ideal conditions. The frequency range used for this comparison was not the range used by the systems but the 4m and 2m bands that are used by BOS. The results are shown in Table 4.

Table 5 shows the systems after standardising them all to a transmit power of 5W.

3.3.1. Interpretation

The translation of the frequency ranges used by BOS shows the DECT standard cannot be compared because its bandwidth requirements are too large. GSM would be possible, however the number of channels is very small and one GSM channel would use the worst frequency.

In systems like TETRA, Tetrapol, DIIS and GSM it can be seen in Table 3 that compared with the values in Table 2 that a substantially higher range is to be expected with these systems on the bad frequency ranges. The systems are easier to compare if the transmitting power is standardised. For this standardisation an output of 5W was selected because hand held radios are specified with a maximum transmitting power of 5W. This comparison of the systems on the same frequency and same transmitting power are shown in Table 5. In this comparison the FM voice transmission is clearly the best for range.

The conclusion from this analysis is that systems that have many decades of tech-

Table 4: System comparison through frequency bands used by BOS.						
	DECT	FM speech	TETRA	Tetrapol	DIIS	GSM GSM-R GSM-BOS
2m band (150MHz) max. bandwidths 0.48MHz & 1.82MHz	Not possible bandwith too large	25 & 92 channel	Possible 19 & 72 channel	Possible 38 & 145 channel	Possible 38 &145 channel	Possible 2 & 9 channel
Range in free space km	126.12	6328.56	582.62	1254.02	2668.73	518.23
4m band (75MHz) bandwidth 3.26MHz	May be possible no more than 1 channel	164 channel	Possible 130 channel	Possible 260 channel	Possible 260 channel	Possible max 16 channel
Range in free space km	252.24	12657.13	1165.24	2508.04	533.47	1036.45

nological development are to be preferred. In addition similar radio connections do not fail abruptly like digital transmissions at a given Bit Error Rate threshold.

Table 5: System comparison for the same frequency and same transmitting power.						
	DECT	FM speech	TETRA	Tetrapol	DIIS	GSM GSM-R GSM-BOS
2m band (150MHz) max. bandwidths 0.48MHz & 1.82MHz	Not possible bandwith too large	25 & 92 channel	Possible 19 & 72 channel	Possible 38 & 145 channel	Possible 38 &145 channel	Possible 2 & 9 channel
Range in free space km	564.03	6328.56	1302.78	2804.08	2668.73	819.39
4m band (75MHz) bandwidth 3.26MHz	May be possible no more than 1 channel	164 channel	Possible 130 channel	Possible 260 channel	Possible 260 channel	Possible max 16 channel
Range in free space km	1128.07	12657.13	2605.56	5608.16	533.47	1638.78

4.

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